

IN THE CLAIMS:

1. (ORIGINAL) A method for ensuring the adequacy of transmission capacity in a digital packet-switched cellular network,
where both voice sample packets and associated header fields are transmitted in real time in one and the same transmission channel,
in which method, if a combined bit count in the voice sample of a packet and in the header field is estimated to exceed the available transmission capacity of the transmission channel, the number of bits in the voice sample is reduced or at least one whole voice block is stolen and the saved number of bits is used for transmitting the header field data of the same packet.
2. (ORIGINAL) The method according to claim 1 wherein the reduction of the number of bits in the voice sample is performed only for packets transmitted at the beginning of a speech spurt.
3. (ORIGINAL) The method according to claim 2 wherein a voice sample replacement is performed when no more than 500 ms have passed from the first VAD included in the same speech spurt.
4. (ORIGINAL) The method according to claim 1 wherein the reduction of the number of bits in the voice sample is performed by replacing the contents of the voice packet with a NO_DATA block.
5. (ORIGINAL) A terminal in a digital packet-switched cellular network which comprises a means for reducing the number of bits in a voice sample included in a packet to be transmitted and a means for using said saved bits for transmitting header field data of the same packet.
6. (ORIGINAL) The terminal according to claim 5 wherein the means for reducing the number of bits in a voice sample included in a packet to be transmitted and means for using said saved bits for transmitting header field data of the same packet comprise:
 - a voice coder for converting a voice sample into a bit combination and for producing a VAD indication,
 - a bit rate and frame count calculation block for calculating the combined bit count for bits in the bit combination transmitted in a packet and bits in the header field after a VAD indication,

- a frame stealing decision block for making a frame stealing decision based on the calculation result from the bit rate and frame count calculation block, and
- a RTP block generation and frame stealing block for replacing in a packet to be transmitted, subsequent to the frame stealing decision, bits in the bit combination produced from the voice sample.

7. (ORIGINAL) The terminal according to claim 5 which comprises a means for reducing the number of bits in a voice sample only for packets transmitted at the beginning of a speech spurt.

8. (ORIGINAL) The terminal according to claim 7 wherein the means for reducing the number of bits in a voice sample are arranged so as to perform a replacement when no more than 500 ms have passed from the first VAD included in the same speech spurt.

9. (ORIGINAL) The terminal according to claim 5 wherein the means for reducing the number of bits in a voice sample, the bit rate and frame count calculation block is arranged so as to replace the contents of the voice packet with a NO_DATA block.

10. (CURRENTLY AMENDED) Software means stored on a terminal in a cellular network which software means is arranged so as to implement the method steps according to claims 1 to 5.

11. (ORIGINAL) Software means according to claim 10, stored on a data storage medium to be loaded into an appropriate cellular network terminal.